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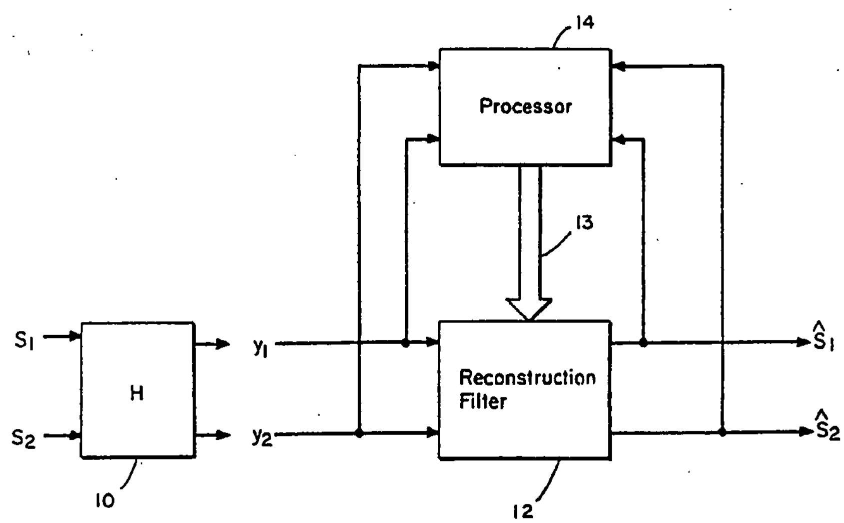

(57) Abstract

A system separates unknown signals which have been combined together through unknown linear filters and for which observations at multiple sensors are made. In a two channel circuit with two inputs and two sensors, the reconstructed source signals are assumed to be decorrelated such that the cross-correlation between the reconstructed source signals is near zero. The transfer functions which represent the crosstalk processes are estimated. The output signals are detected and the transfer functions are recursively solved. A reconstruction filter is used to recover the original input signals.

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MULTI-CHANNEL SIGNAL SEPARATION

Background of the Invention

In a variety of contexts, observations are made of the outputs of an unknown multiple-input multiple-output linear system, from which it is of interest to recover the input signals. For example, in problems of enhancing speech in the presence of background noise, or separating competing speakers, multiple microphone measurements will typically have components from both sources with the linear system representing the effect of the acoustic environment.

Considering specifically the two-channel case, it is desired to estimate the two source signals s_1 and s_2, from the observation of the two output signals y_1 and y_2. In many applications one of the signals, s_1, is the desired signal, while the other signal, s_2, is the interference or noise signal. Both desired and noise signals are coupled through the unknown system to form the observed signals.

The most widely used approach to noise cancellation, in the two channel case, was suggested by Widrow et al. in "Adaptive Noise Canceling: Principles and Applications", Proc. IEEE,

63:1692-1716, which was published in 1975. In this approach, it is assumed that one of the observed signals, y_1 - the primary signal, contains both the desired signal and an uncorrelated interfering signal, while the other observed signal, y_2 - the reference signal, contains only the interference. The system that couples the reference signal into the primary signal is found using the least mean square (LMS) algorithm. Then, the reference signal is filtered by the estimated system and subtracted from the primary signal to yield an estimate of the desired signal, s_1. For example, U.S. Patent No. 4,773,906 issued to Warnaka et al. used Widrow's approach and assumptions in an acoustic cancellation structure.

The main drawback of Widrow's approach lies in the crucial assumption that the reference signal is uncorrelated with the primary signal. This assumption is not realized in practice due to leakage of the primary signal into the reference signal. This degrades the performance of Widrow's method. Depending on the leakage (or cross-talk) power, this degradation may be severe, leading to a reverberant quality in the reconstructed signal since a portion of the desired signal is also subtracted, out of phase, together with the interference signal. Thus, a method of reconstruction is needed that can operate without making the unrealistic no-leakage assumption.

Summary of the Invention

The invention proposes an approach to estimate a plurality of input source signals, which passed through an unknown multiple-input multiple-output system, from observations of the output signals. It suggests, in fact, a solution of the problem of signal separation from their coupled observations. In the two channel case, the invention suggests an approach to estimate the desired signal (i.e., to cancel the interfering signal) or to estimate both source signals, in the more realistic scenario where both signals are coupled into the observed signals, and thereby overcoming the main drawback of the previous techniques.

The invention comprises a signal processing system including a plurality of detectors for receiving plural observed signals, which, as mentioned above, results from plural source signals subjected to an unknown transfer function. A processor receives the observed signals and estimates the components of the transfer function, which are used to produce a filter (for example, but not necessarily, an inverse filter) for reconstructing the input source signals. The criterion for the estimate, or for the production of the reconstruction filter, is that the reconstructed source signals are uncorrelated. No assumption is made about the observed signals. Decorrelating

the reconstructed signals implies that the source signals, which became correlated in the observation due to coupling, are separated at reconstruction.

In the two channel case, the original source signals are denoted s_a and s_b, the separation system will only reconstruct s_1 and s_2, representing the separated source signals processed by the filters. In this two channel case, when the reconstructing filter is an inverse filter, i.e., its response is the inverse of the system response, the first observed signal, y_1, is passed through a filter $\hat{H}_{21}$ (which is an estimate of the system H_{21} that couples s_1 into y_2), to yield a signal v_1. The other observed signal y_2 is passed through a filter $\hat{H}_{12}$ (which is an estimate of the system H_{12} that couples s_2 into y_1), to yield a signal v_2. Then v_1 is subtracted from y_2, v_2 is subtracted from y_1 and each of the resulting difference signals is passed through another system 1/G, to yield the reconstructed source signals $\hat{s}_1$ and $\hat{s}_2$. The components of the reconstruction filter, $\hat{H}_{12}$, $\hat{H}_{21}$ and 1/G are estimated from the observed data so that the reconstructed signals are decorrelated. When it is assumed that the coupling system H_{21} is zero, and its estimate is forced to be zero, the standard Widrow's method results. By assuming decorrelated reconstructed outputs, no assumptions need be made regarding the observed signal. If one transfer component of the reconstruction filter is known,

then the other transfer component can be estimated. This criterion can be used with several other assumptions. For example, the generating system transfer components H_{12} and H_{21} can have finite impulse responses which have finite lengths. Then, the inverse filter for the generating system can be calculated to determine the reconstruction filter.

In another embodiment, the reconstruction filter can have a finite impulse response. If either H_{21} or H_{12} are known, then the other can be calculated and reconstructed output signals developed. Also, only partial knowledge of the transfer components is needed to accurately calculate the reconstruction filter.

In the preferred embodiment, it is assumed that the generating system transfer function is an FIR filter; thus, the coupling systems H_{12} and H_{21} have a finite impulse response (FIR). In other words, the coupling filters can be represented as tapped delay lines, where each delay segment is multiplied by a filter coefficient and all the outcomes are summed to yield the filter output. The estimate of the coupling FIR system is performed via an iterative process. The processor alternately solves for the coefficients of one system, assuming that the other system is known.

A similar processor and embodiment is obtained when the reconstruction filter is forced to be an FIR system.

In another related embodiment, the processor that determines the unknown transfer functions and produces the reconstruction filter (e.g., the inverse filter) is adaptive. The processor updates the components of the transfer function as more data is available, and based on the previous estimate of the input source signals. This embodiment represents a class of possible algorithms of various methods for sequential updating of the parameters.

One application of the invention would be in a signal enhancing scenario in a room. The first (primary) microphone is close to the speaker and the other (reference) microphone is close to the noise source. However, the microphones need not be close to the sources. The first microphone detects the desired speech signal with some noise, and the second microphone detects mainly the noise with some speech leakage. The coupling is due to the unknown acoustic room environment. The processor of the invention is used to reconstruct the desired speech signal without the interfering noise. When the second signal source is another speaker, the processor of the invention separates both speech signals from the detected microphone signals that contain their coupled measurements. In the context of speech signals, the invention processor, for example, can be used as a front-end in automatic speech recognition systems operating in noisy environments, or when there is a cross-talk from an interfering speaker.

Although the preferred embodiment operates for speech signal in an acoustic environment, the system for multi-channel signal separation can be applied to many different areas. For example, a 60 Hz interference from electrocardiogram can be deleted, even in cases when the reference contains some of the desired signal. Other applications may include cancellation of noise signals in telemetry system. Also, interfering signals in an antenna system can be cancelled using the proposed algorithm.

The above and other features of the invention including various novel details of construction and combinations of parts will now be more particularly described with reference to the accompanying drawings and pointed out in the claims. It will be understood that the particular choice embodying the invention is shown by way of illustration only and not as a limitation of the invention. The principles and features of this invention may be employed in various and numerous embodiments without departing from the scope of the invention.

Brief Description of the Drawings

Figure 1 is a schematic illustration of a general signal processing system embodying the present invention.

Figure 2 illustrates a general two channel system where H_{11}, H_{12}, H_{21} and H_{22} represent the transfer functions of four single-input single-output systems.

Figure 3 represents a schematic representation of H^{-1}, the inverse of the estimated system of Figure 2.

Figure 4 represents another schematic representation of a possible reconstruction system.

Figure 5 illustrates an embodiment of the signal processing system that implements an iterative process for estimating the coupling systems.

Figure 6 illustrates an embodiment of the signal processing system that implements an adaptive sequential algorithm for estimating the coupling systems.

Figure 7 illustrates a speech enhancing and noise cancellation system using the signal processing circuit of the invention.

Figure 8 illustrates a signal enhancing system in an underwater acoustic environment.

Figure 9 illustrates a tapped delay line.

Detailed Description of the Invention

Figure 1 illustrates schematically the operation of the signal separator of the invention. Two signals, s_1 and s_2, are processed by an environment 10 which is represented by a transfer function H. Figure 2 shows the transfer function H in more detail. The transfer function H produces output signals y_1 and y_2. The output signal y_1 comprises the signal s_1 and a filtered version of signal s_2. The output signal y_2 comprises the

signal s_2 and a filtered version of signal s_1. Thus, the environment 10 represented by transfer function H produces signal leakage or cross talk in both output signals y_1 and y_2. The processor 14 and the reconstruction filter 12 operate under the criterion that the reconstructed source signals, $\hat{s}_1$ and $\hat{s}_2$, are uncorrelated.

Processor 14 recursively operates on the values of detected output signals, y_1 and y_2, and reconstructed source signals, s_1 and s_2, to produce an output 13 which updates the system components of reconstruction filter 12. Thus, crosstalk in the detected output signals, y_1 and y_2, is compensated and the reconstructed source signals, s_1 and s_2, converge to the actual values of signals s_1 and s_2, respectively. A detailed explanation of this process is recited below.

A general two-channel linear time invariant (LTI) system can be represented in the form shown in Figure 2, where $H_{11}(w)$, $H_{12}(w)$, $H_{21}(w)$, and $H_{22}(w)$ represent the transfer functions of four single-input single-output LTI systems. Thus, Figure 2 illustrates the environment of $H(w)$ that is used to develop the inverse transfer function $H^{-1}(w)$ which is required to reconstruct signal sources $s_1(n)$ and $s_2(n)$. The problem specifically considers estimating $s_1[n]$, $s_2[n]$, $H_{12}(\omega)$, and $H_{21}(\omega)$. In some scenarios, such as when one sensor is close to each of the signal sources $s_a[n]$ and $s_b[n]$, it is appropriate to assume $H_{11}(w) = H_{22}(w) = 1$. In other

scenarios, when this is not the case, then the formulation is still directed at separation of $s_1[n]$ and $s_2[n]$, but not full recovery of the input signals. In many cases this may still be an acceptable objective. For example, if either $s_a[n]$ and $s_b[n]$ or both are speech, for a wide class of transfer functions $H_{11}(w)$ and $H_{22}(w)$ separation of $s_1[n]$ and $s_2[n]$ from $y_1[n]$ and $y_2[n]$ is sufficient for intelligibility. Also, assume that $H_{12}(w)$ and $H_{21}(w)$ are stable LTI systems, and denote their unit sample responses by $h_{12}[n]$ and $h_{21}[n]$ respectively. Consequently,

(1)

$$y_1[n] = s_1[n] + h_{12}[n] * s_2[n]$$
$$y_2[n] = s_2[n] + h_{21}[n] * s_1[n] \qquad (2)$$

where * denotes convolution. We further assume that

$$H_{12}(\omega)H_{21}(\omega) \neq 1 \qquad \forall -\pi < \omega < \pi \qquad (3)$$

The signals $s_1[n]$ and $s_2[n]$ are considered to be sample functions from stochastic processes having stationary covariance functions. For notational simplicity, it is assumed that $s_1[n]$ and $s_2[n]$ are zero mean. Consequently the auto-covariances and cross-covariances are given by

expectation

$$r_{11}[k] \triangleq E\{s_1[n]s_1^*[n-k]\} \qquad (4a)$$

expectation

$$r_{22}[k] \triangleq E\{s_2[n]s_2^*[n-k]\} \tag{4b}$$

$$r_{12}[k] \triangleq E\{s_1[n]s_2^*[n-k]\} \tag{4c}$$

where E(.) denotes expected value. The corresponding power spectra will be denoted as $P_{s1s1}(\omega)$, $P_{s2s2}(\omega)$, and $P_{s1s2}(\omega)$ respectively. It should be stressed that the zero mean assumption is not necessary and the derivation and results apply equally to the more general case of non-zero and time-varying mean values, since they are phrased in terms of covariances.

The desired approach comprises developing estimates of $s_1[n]$, $s_2[n]$, and the linear systems $H_{12}(\omega)$ and $H_{21}(\omega)$ subject to satisfying the model of Equations (1) and (2), the additional assumption of Equation(3), and appropriate constraints on $P_{s1s1}(\omega)$, $P_{s2s2}(\omega)$, and $P_{s1s2}(\omega)$.

From Equations (1) and (2) or Figure 2, the transfer function matrix H(w) is

$$\boldsymbol{H}(\omega) = \begin{bmatrix} 1 & H_{12}(\omega) \\ H_{21}(\omega) & 1 \end{bmatrix} \tag{5}$$

and correspondingly the transfer function of the inverse system is

$$\boldsymbol{H}^{-1}(\omega) = \frac{1}{1 - H_{12}(\omega)H_{21}(\omega)} \begin{bmatrix} 1 & -H_{12}(\omega) \\ -H_{21}(\omega) & 1 \end{bmatrix} \tag{6}$$

which is stable under the condition of Equation (3).

From (6) and the well-known relationship for power spectra at the input and output of LTI systems, we can write that

$$|1-\hat{H}_{12}(\omega)\hat{H}_{21}(\omega)|^2 P_{\hat{s}_1\hat{s}_2}(\omega) = P_{y_1y_2}(\omega) - \hat{H}_{12}(\omega)P_{y_2y_2}(\omega) - \hat{H}^{*}_{21}(\omega)P_{y_1y_1}(\omega) + \hat{H}_{12}\hat{H}^{*}_{21}(\omega)P^{*}_{y_1y_2}(\omega) \quad (7a)$$

$$|1-\hat{H}_{12}(\omega)\hat{H}_{21}(\omega)|^2 P_{\hat{s}_1\hat{s}_1}(\omega) = P_{y_1y_1}(\omega) - \hat{H}_{12}(\omega)P_{y_2y_1}(\omega) - \hat{H}^{*}_{12}(\omega)P_{y_1y_2}(\omega) + \hat{H}_{12}(\omega)\hat{H}^{*}_{12}(\omega)P_{y_2y_2}(\omega) \quad (7b)$$

$$|1-\hat{H}_{12}(\omega)\hat{H}_{21}(\omega)|^2 P_{\hat{s}_2\hat{s}_2}(\omega) = P_{y_2y_2}(\omega) - \hat{H}_{21}(\omega)P_{y_1y_2}(\omega) - \hat{H}^{*}_{21}(\omega)P_{y_2y_1}(\omega) + \hat{H}_{21}(\omega)\hat{H}^{*}_{21}(\omega)P_{y_1y_1}(\omega) \quad (7c)$$

If $P_{s1s1}(\omega)$, $P_{s2s2}(\omega)$, and $P_{s1s2}(\omega)$ are known, estimates of $H_{12}(\omega)$, $H_{21}(\omega)$, $s_1[n]$, and $s_2[n]$ consistent with Equations (7) and the model of Equation (1) are determined. However, the source power spectra are generally not known, so Equation (7) has too many variables to be solved.

I. Estimation Based on Decorrelation

It has been determined that Equation 7(a) can be readily solved by assuming that $P_{s1s2}(\omega) = 0$. By assuming that $P_{s1s2}(\omega) = 0$, i.e. that s1 and s2 are not correlated, either H_{12} or H_{21} can be determined in terms of the power spectra for detected signals y1 and y2 and the estimate for the other transfer function component exists. Thus, an adequate number

of unknowns are eliminated from Equation 7(a) to permit its solution.

Estimates of $s_1[n]$, $s_2[n]$ and the linear systems $H_{12}(\omega)$ and $H_{21}(\omega)$ to satisfy the model of Equations (1) and (2) and the constraint that $P_{s1s2}(\omega) = 0$ are determined recursively. Specifically, $H_{12}(\omega)$ and $H_{21}(\omega)$, estimates of $H_{12}(\omega)$ and $H_{12}(\omega)$, are chosen to satisfy (7a) with $P_{s1s2}(\omega) = 0$, i.e.

$$P_{y_1y_2}(\omega) - \hat{H}_{12}(\omega)P_{y_2y_2}(\omega) - \hat{H}^*_{21}(\omega)P_{y_1y_1}(\omega) + \hat{H}_{12}(\omega)\hat{H}^*_{21}(\omega)P^*_{y_1y_2}(\omega) = 0 \qquad (8)$$

where $P_{y1y2}(\omega)$, $P_{y2y2}(\omega)$, and $P_{y1y1}(\omega)$ are estimated from the observed data.

The estimates $s_1[n]$ and $s_2[n]$ are then obtained by applying to the data the inverse filter $H^{-1}(\omega)$ given by

$$\hat{H}^{-1}(\omega) = \frac{1}{1 - \hat{H}_{12}(\omega)\hat{H}_{21}(\omega)} \begin{bmatrix} 1 & -\hat{H}_{12}(\omega) \\ -\hat{H}_{21}(\omega) & 1 \end{bmatrix} \qquad (9)$$

Referring to the embodiment of Figure 1, processor 14 estimates a first system cross component, generates the second system cross component to satisfy Equation (8), and then s_1 and s_2 are generated by the reconstruction filter, using an inverse filter.

Clearly, Equation (8) does not specify a unique solution for both $H_{12}(\omega)$ and $H_{21}(\omega)$ and any solution will result in estimates of $s_1[n]$ and $s_2[n]$ which satisfy the decorrelation condition. Several specific approaches which further constrain the problem to eliminate this ambiguity are discussed in Section II.

If one of the transfer function components $H_{12}(\omega)$, $H_{21}(\omega)$ is known, the other can be solved for easily using Equation (8). For example, one straightforward constraint which leads to a unique solution of (8) and unique estimates of $s_1[n]$ and $s_2[n]$ results from having $H_{21}(\omega)$ specified. In this case,

$$\hat{H}_{12}(\omega) = \frac{\hat{P}_{y_1 y_2}(\omega) - H_{21}^{*}(\omega)\hat{P}_{y_1 y_1}(\omega)}{\hat{P}_{y_2 y_2}(\omega) - H_{21}^{*}(\omega)\hat{P}_{y_1 y_2}^{*}(\omega)} \qquad (10)$$

If the data satisfies the model of Figure 1 and if the estimates of $P_{y1y2}(\omega)$ and P_{y1y2} are exact, then the estimate of $H_{12}(\omega)$ will also be exact.

Correspondingly, if $H_{12}(\omega)$ is known, then

$$\hat{H}_{21}(\omega) = \frac{\hat{P}_{y_2 y_1}(\omega) - H_{12}^{*}(\omega)\hat{P}_{y_2 y_2}(\omega)}{\hat{P}_{y_1 y_1}(\omega) - H_{12}^{*}(\omega)\hat{P}_{y_1 y_2}^{*}(\omega)} \qquad (11)$$

Thus, processor 14 of Figure 1 can utilize either Equation (10) or (11), where one transfer

function component is known, to find the other component, and apply the inverse filter to generate the source signals.

When $H_{21}(\omega)$ equals zero, the solution reduces to the result used in the least mean square (LMS) calculation. As an interesting special case, suppose $H_{21}(\omega)$ is assumed to be zero, i.e. in Figure 2 there is assumed to be no coupling from $s_2[n]$ to the first output $y_1[n]$. In this case, from (10),

$$\hat{H}_{12}(\omega) = \frac{P_{y_1 y_2}(\omega)}{P_{y_2 y_2}(\omega)} \tag{12}$$

and

$$\hat{s}_1[n] = y_1[n] - \hat{h}_{12}[n] * y_2[n] \tag{13}$$

This choice of $H_{12}(\omega)$ is in fact identical to that used in Widrow's least squares approach. Specifically, it is straightforward to show that if $s_1[n]$ is estimated according to (13), the choice of $H_{12}(\omega)$ as specified in (12) results in $E(s_1^2[n])$ being minimized. The least squares approach of Widrow has been extremely successful in a wide variety of contexts. It is also generally understood that some of its limitations derive from the key assumption that $H_{12}(\omega) = 0$. Equation (11) suggests a potentially interesting modification of

the least squares method to incorporate non-zero but specified coupling.

Figure 3 illustrates the inverse of the estimated system of Figure 2. Reconstruction filter 12 uses an inverse filter to reconstruct source signals s_1 and s_2. Detected first observed signal, y_1, is passed through system H_{12} to yield U_2. The other observed signal y_2 is passed through a filter H_{21} to yield a signal U_1. Then, U_2 is subtracted from y_2, U_2 is subtracted from y_1, and each of the resulting difference signals which are V_1 and V_2, respectively, are passed through another system, 1/G to yield the reconstructed source signals s_1 and s_2. If the original system is not comprised of finite impulse response filters FIRs, both system components cannot be determined. One component must be assumed and the other solved.

Figure 4 illustrates a reconstruction filter where the cross transform components, 38 and 40, are FIRs. As will be discussed in detail below, this permits the concurrent calculation of both cross components.

If both $H_{12}(\omega)$ and $H_{21}(\omega)$ are unknown, we then need to jointly estimate them. As already indicated, imposing the assumption that the signals are uncorrelated is insufficient to uniquely solve the problem. In order to obtain a unique solution, we must incorporate additional assumptions concerning the underlying model.

One possibility is to assume statistical independence between $s_1[n]$ and $s_2[n]$. If the signals are not jointly Guassian, this is a stronger condition than the assumption that the signals are statistically uncorrelated. By imposing statistical independence between the estimated signals, additional constraints on the high order moments of $s_1[n]$ and $s_2[n]$ are obtained, that can be used to specify a unique solution for both filters.

II. Estimation Based on Decorrelation and an FIR Constraint

Another class of constraints leading to a unique solution for $H_{12}(\omega)$ and $H_{21}(\omega)$ is comprised of restricting the two linear filters to be causal and finite impulse response (FIR) with a pre-specified order.

As explained below, both h_{12} and h_{21} can be represented as tapped delay lines with filter coefficients a_n and b_n, respectively. Figure 9 illustrates a tapped delay line which can be used as an FIR filter. The line consists of a plurality of delays 90, 92, 94, 96, and 98. The output of each delay is multipled by a respective coefficient a1 to a5. This multiplication occurs in multipliers 102, 104, 106, 108, and 110. The outputs of the multipliers are added by summer 100 to produce an output signal y.

Consider $h_{12}[n]$ and $h_{21}[n]$ to be FIR filters and, thus, of the form

$$h_{12}[n] = \begin{cases} a_n & 0 \le n \le q_1 - 1 \\ 0 & \text{otherwise} \end{cases} \quad (14)$$

$$h_{21}[n] = \begin{cases} b_n & 0 \le n \le q_2 - 1 \\ 0 & \text{otherwise} \end{cases} \quad (15)$$

The filter coefficients $\{a_n\}$ and $\{b_n\}$ are unknown and the filter lengths q_1 and q_2 are assumed known. Since we now have (q_1+q_2) unknowns, Equation (8) or its equivalent in the time domain should, in principle, provide a sufficient number of (nonlinear) equations in the unknown filter coefficients to obtain a unique solution. In the following discussion, a specific time domain approach to obtaining the filter coefficients is developed. As noted above, the inverse system $H^{-1}(w)$ as given by Equation (6) is represented by the block diagram in Figure 3, where $G(w) = 1 - H_{12}(w)H_{21}(w)$. The Figure 3 parameters, $v_1(n)$, $v_2(n)$, and g_n, are defined as follows:

$$\begin{aligned} \hat{v}_1(n) &= y_1(n) - \hat{h}_{12}(n) \circ y_2(n) \\ &= y_1(n) - \hat{a}_n \circ y_2(n) \end{aligned} \quad (16)$$

$$\begin{aligned} \hat{v}_2(n) &= y_2(n) - \hat{h}_{21}(n) * y_1(n) \\ &= y_2(n) - \hat{b}_n * y_1(n) \end{aligned} \qquad (17)$$

and

$$\begin{aligned} \hat{g}_n &= \delta_n - \hat{h}_{12}(n) * \hat{h}_{21}(n) \\ &= \delta_n - \hat{a}_n * \hat{b}_n \end{aligned} \qquad (18)$$

where δ_n is the Kroneker delta function. With these definitions, the equations for the signal estimates are given by:

$$\hat{g}_n * \hat{s}_1(n) = \hat{v}_1(n) \qquad (19)$$

$$\hat{g}_n * \hat{s}_2(n) = \hat{v}_2(n) \qquad (20)$$

From Figure 3 (or Equations (16) and (17)), Equation (8) can be rewritten in either of the following two forms:

$$\hat{H}_{12}(\omega) = \frac{P_{y_1 v_2}(\omega)}{P_{y_2 v_2}(\omega)} \qquad (21)$$

$$\hat{H}_{21}(\omega) = \frac{P_{y_2 v_1}(\omega)}{P_{y_1 v_1}(\omega)} \qquad (22)$$

where $P_{yivj}(\omega)$ i,j = 1,2 is the cross spectrum between $y_i(n)$ and $v_j(n)$.

Equivalently in the time (lag) domain we obtain:

$$\hat{a}_n * c_{y_2v_2}(n) = c_{y_1v_2}(n) \tag{23}$$

$$\hat{b}_n * c_{y_1v_1}(n) = c_{y_2v_1}(n) \tag{24}$$

where $c_{yivj}(k)$ is the covariance function between $y_i(n)$ and $v_j(n)$, i.e.,

$$c_{y_iv_j}(n) = E\{y_i(k)v_j^+(k-n)\} - E\{y_i(k)\}E^+\{v_j(k-n)\} \qquad i,j = 1,2 \tag{25}$$

and $^+$ denotes complex conjugation.

Under the assumption that $s_i(t)$ i = 1,2 are zero mean then $y_i(t)$ and $v_j(t)$ i,j = 1/2 are also zero mean, in which case the second term on the right side of Equation (30) to be discussed below, equals zero, and $c_{yivj}(n)$ is a correlation function. Otherwise, the covariance must be computed.

Expressing (23) for n = 0,1,2....,(q_1-1) in a vector form

$$C_{y_2v_2}\left(\hat{b}\right)\hat{a} = c_{y_1v_2}\left(\hat{b}\right) \tag{26}$$

where

$$\hat{a} = \begin{bmatrix} \hat{a}_0 \\ \hat{a}_1 \\ \vdots \\ \hat{a}_{q_1-1} \end{bmatrix} \qquad (27)$$

$$\hat{b} = \begin{bmatrix} \hat{b}_0 \\ \hat{b}_1 \\ \vdots \\ \hat{b}_{q_2-1} \end{bmatrix} \qquad (28)$$

Similarly, expressing (24) for n = 1/2/...,(q_2-1) in vector form

$$C_{y_1 v_1}(\hat{a})\hat{b} = c_{y_2 v_1}(\hat{a}) \qquad (29)$$

Thus, for a pre-specified b, the solution to a is given by

$$\hat{a} = C^{-1}_{y_2 v_2}\left(\hat{b}\right) c_{y_1 v_2}\left(\hat{b}\right) \qquad (30)$$

and for a pre-specified a, the solution to b is given by

$$\hat{b} = C^{-1}_{y_1 v_1}(\hat{a}) c_{y_2 v_1}(\hat{a}) \qquad (31)$$

If b = 0 is substituted in (30), we obtain the least squares estimate of a (under the assumption that in fact b = 0). Equation (30) therefore

suggests a modification to the least squares solution in the FIR case, that incorporates situations in which b is non-zero.

To estimate a or b from the observed data, we replace the covariances in Equations (30) and (31) with their sample estimates. Specifically, the expectations are approximated by the empirical averages:

$$C_{y_2 v_2}\left(\hat{b}\right) \approx \frac{1}{\sum_{k=1}^{n} \beta^{n-k}} \sum_{k=1}^{n} \beta^{n-k} v_2^{+}(k) y_2^{\top}(k) \tag{32}$$

$$c_{y_1 v_2}\left(\hat{b}\right) \approx \frac{1}{\sum_{k=1}^{n} \beta^{n-k}} \sum_{k=1}^{n} \beta^{n-k} v_2^{+}(k) y_1^{\top}(k) \tag{33}$$

$$C_{y_1 v_1}(\hat{a}) \approx \frac{1}{\sum_{k=1}^{n} \alpha^{n-k}} \sum_{k=1}^{n} \alpha^{n-k} v_1^{+}(k) y_1(k) \tag{34}$$

$$c_{y_2 v_1}(\hat{a}) \approx \frac{1}{\sum_{k=1}^{n} \alpha^{n-k}} \sum_{k=1}^{n} \alpha^{n-k} v_1^{+}(k) y_2(k) \tag{35}$$

where α and β are real numbers between 0 and 1. To achieve maximal statistical stability, we choose $\alpha = \beta = 1$. If, however, the signals and/or the unknown parameters exhibit non-stationary behavior in time, it is preferable to choose α, $\beta < 1$. In this way, exponential weighting can be introduced so that the signal and parameter estimates depend more heavily

on the current data samples, and in effect an adaptive algorithm is created that is capable of tracking the time-varying characteristics of the underlying system.

The form of the empirical averages in (32) and (33) suggests solving (30) recursively. Using

$$
\begin{aligned}
Q(n) &= \left[\sum_{k=1}^{n} \beta^{n-k} v_2^{+}(k) y_2^{T}(k)\right]^{-1} \qquad (36)\\
&= \left[\beta Q^{-1}(n-1) + v_2^{+}(n) y_2^{T}(n)\right]^{-1}\\
&= \frac{1}{\beta}\left[Q(n-1) - \frac{Q(n-1) v_2^{+}(n) y_2^{T}(n) Q(n-1)}{\beta + y_2^{T}(n) Q(n-1) v_2^{+}(n)}\right]
\end{aligned}
$$

and

$$
\begin{aligned}
q(n) &= \sum_{k=1}^{n} \beta^{n-k} v_2^{+}(k) y_1(k)\\
&= \beta q(n-1) + v_2^{+}(n) y_1(n) \qquad (37)
\end{aligned}
$$

a(n), the estimate of a based on data to time n, can be defined as

$$
\begin{aligned}
\hat{a}(n) &= Q(n) q(n)\\
&= Q(n)\left[\beta q(n-1) + v_2^{+}(n) y_1(n)\right]\\
&= Q(n)\left[\beta Q^{-1}(n-1)\hat{a}(n-1) + v_2^{+}(n) y_1(n)\right]\\
&= Q(n)\{\beta \cdot \frac{1}{\beta}[Q^{-1}(n) - v_2^{+}(n) y_2^{T}(n)]\hat{a}(n-1) + v_2^{+}(n) y_1(n)\}\\
&= \hat{a}(n-1) + Q(n)[v_2^{+}(n) y_1(n) - v_2^{+}(n) y_2^{T}(n)\hat{a}(n-1)]\\
&= \hat{a}(n-1) + Q(n) v_2^{+}(n)[y_1(n) - y_2^{T}(n)\hat{a}(n-1)]\\
&= \hat{a}(n-1) + Q(n) v_2^{+}(n) v_1(n|n-1) \qquad (38)
\end{aligned}
$$

where

$$v_1(n|n-1) = y_1(n) - y_2^T(n)\hat{a}(n-1) \quad (39)$$

Similarly, the solution to (31) can be computed recursively as follows:

$$\hat{b}(n) = \hat{b}(n-1) + R(n)v_1^+(n)v_2(n|n-1) \quad (40)$$

$$R(n) = \frac{1}{\alpha}\left[R(n-1) - \frac{R(n-1)v_1^+(n)y_1^T(n)R(n-1)}{\alpha + y_1^T(n)R(n-1)v_1^+(n)}\right] \quad (41)$$

where

$$R(n) = \left[\sum_{k=1}^{n} \beta^{n-k} v_1^+(k) y_1^T(k)\right]^{-1} \quad (42a)$$

and

$$v_2(n|n-1) = y_2(n) - y_1^T(n)\hat{b}(n-1) \quad (42b)$$

If b = 0, then (38) becomes the well-known recursive least squares (RLS) solution of the least squares problem. Thus, (38) can be viewed as a generalization of the RLS solution to a when b is specified and not restricted to be zero. Similarly, (40) is a generalized RLS algorithm for obtaining b when a is specified and not restricted to be zero.

Time domain solutions can also be adequately handled by tapped delay line methods.

At times, both a and b are unknown vector parameters to be estimated. The form of (30) and (31) suggests the following iterative algorithm:

$$\hat{a}^{(\ell)} = C^{-1}_{y_2 v_2}\left(\hat{b}^{(\ell-1)}\right) c_{y_1 v_2}\left(\hat{b}^{(\ell-1)}\right) \tag{43}$$

$$\hat{b}^{(\ell)} = C^{-1}_{y_1 v_1}\left(\hat{a}^{(\ell-1)}\right) c_{y_2 v_1}\left(\hat{a}^{(\ell-1)}\right) \tag{44}$$

where $a^{(\ell)}$ and $b^{(\ell)}$ are, respectively, the estimates of a and b after ℓ iteration cycles. To implement the algorithm, we use the empirical averages given in Equations (32)-(35). Of course, for a given $b^{(\ell-1)}$, $a^{(\ell)}$ may be computed recursively in time using (38). Similarly, for a given $a^{(\ell-1)}$, $b^{(\ell)}$ may be computed using (40).

This iterative process is shown in the embodiment of Figure 5. To obtain a fully recursive algorithm, we suggest incorporating (38) with (39), where in the recursion for estimating a, b(n) is substituted for $b^{(\ell)}$, and in the recursion for estimating b, a(n) is substituted for $a^{(\ell)}$. Note that replacing the iteration index by the time index is a common procedure in stochastic approximation.

Thus, as an alternative to (43) and (44), the following iterative procedure is used to solve (26) and (29):

$$\hat{a}^{(\ell)} = \hat{a}^{(\ell-1)} + \gamma_1^{(\ell)} \left[c_{y_1 v_2}\left(\hat{b}^{(\ell-1)}\right) - c_{y_2 v_2}^{-1}\left(\hat{b}^{(\ell-1)}\right) \hat{a}^{(\ell-1)} \right] \quad (45)$$

$$\hat{b}^{(\ell)} = \hat{b}^{(\ell-1)} + \gamma_2^{(\ell)} \left[c_{y_2 v_1}\left(\hat{a}^{(\ell-1)}\right) - c_{y_1 v_1}^{-1}\left(\hat{a}^{(\ell-1)}\right) \hat{b}^{(\ell-1)} \right] \quad (46)$$

where $\gamma_1^{(\ell)}$ and $\gamma_2^{(\ell)}$ are constant gains (step sizes) that may depend on the iteration index. This algorithm can be viewed as an iterative gradient search algorithm for solving (26) and (29).

Using,

$$c_{y_1 v_2}\left(\hat{b}\right) - C_{y_2 v_2}^{-1}\left(\hat{b}\right) \hat{a} = E\{v_2^{+}(n) v_1(n)\} \quad (47)$$

$$c_{y_2 v_1}(\hat{a}) - C_{y_2 v_2}^{-1}(\hat{a}) \hat{b} = E\{v_1^{+}(n) v_2(n)\} \quad (48)$$

Then, using the first order stochastic approximation method, replace expectation by current realization, and iteration index by the time index to obtain the following sequential algorithm:

$$\hat{a}(n) = \hat{a}(n-1) + \gamma_1(n) \cdot v_2^+(n|n-1)v_1(n|n-1) \quad (49)$$

$$\hat{b}(n) = \hat{b}(n-1) + \gamma_2(n) \cdot v_1^+(n|n-1)v_2(n|n-1) \quad (50)$$

where $v_1^+(n/n-1)$ is the vector defined in (39)

computed using $a = a(n-1)$, and $v_2^+(n/n-1)$ is the vector defined using $b = b(n-1)$.

The flow chart in Figure 5 illustrates the adaptive sequential algorithm which estimates a (i) and b (i) recursively using Equations (49) and (50), then generating v_1 (i) and v_2 (i) according to Equations (16) and (12) and finally generating s_1 and s_2 according to Equations (19) and (20), respectively. Block 110 illustrates the setting of the initial estimate of a. As shown in block 112, v_1 is calculated using the initial a values and the detected signals, y_1 and y_2, in accordance with Equation (16). The value v_1 is used to perform the calculations shown in block 114 in Figure 5(a). According to Equations (34) and (35), values are calculated for c_{y1v1} and c_{y2v1}. Next the values for b are calculated using Equations (31) and (41) as shown in block 116. The values for b are then used to find v_2 in accordance with Equation (17) as shown by block 118. Using this value, Equations (32) and (33) are used to calculate C_{y2v2} and c_{y1v2} as shown in block 120. Next, values are found for $A^{(k+1)}$.

Block 124 determines whether convergence has occurred. If more iterations need to be calculated, k is set equal to k+1 and calculations are restarted in block 112. If conversion has occurred, the values for $a^{(k+1)}$ are set equal to the estimate for H_{12}. Also in block 128, the calculated values for b_i are set equal to the estimate for the transform H_{21}. The final calculations are made as shown in block 130 and 132. As shown in block 130, g_n is calculated using Equation (18) and, finally, estimates are made for the reconstructed output signals s_1 and s_2 using Equations (19) and (20), respectively.

Figure 6 illustrates an embodiment of the signal processing system that implements the adaptive sequential algorithm of the flow chart of Figure 5. Detected signals y_1 and y_2 are delayed. Delay line 44 with a length q_2 delays detected signal y_1 and delay line 46 with the length q_1 delays the detected signal y_2. In block 42, values for v_1 (n/n-1) are calculated in accordance with Equation (39). Blocks (46) and (48) are used to calculate Q(n) in accordance with Equation (36). Values for the estimated filter coefficients a are calculated in accordance with Equation (38) by using the outputs of boxes 42 and 46. The values for a(n), as calculated by the processor of Figure 6, are used to update the reconstruction filter 12 of Figure 1. A similar process is used to calculate the values of filter coefficients b. Thus, updated

reconstructed output signals can be derived. It should be noted that this embodiment represents a class of algorithms where the adaptive techniques can be varied to make a smooth estimate of several components of v_1 and v_2.

By assuming the reconstructed outputs are decorrelated, no assumptions need be made regarding the observed signal. If one transfer component, for example H_{12}, is known, then the other transfer component H_{21} can be estimated. The criterion of decorrelated reconstructed signal outputs can be used with several other assumptions. For example, the generating system transfer components H_{12} and H_{21} can have finite impulse responses. Then the inverse filter for the generating system can be calculated. In another example, the reconstruction filter can have a finite impulse response. If either H_{21} or H_{12} are known, then the other can be calculated and reconstructed output signals developed. Also, only partial knowledge of the transfer components is needed to accurately calculate the reconstruction filter. Thus, if either the phase or magnitude of the filter components is known, the other parameter can be calculated and the reconstruction filter completely described. These examples are not limiting, but merely representative of some of the contexts in which the signal processing device of the invention can be used.

The reconstruction filter with some parameters unknown can also be updated or modified with time variant data by forcing the output to be decorrelated. Thus, the algorithms can be adaptive in time and sequential, i.e. updated as more data is obtained. The algorithms can also be used in concurrent processing.

Figure 7 illustrates the preferred use of the invention in a speech enhancing and noise cancellation system. The separation of voice signals from interfering noise signals in a room can be easily accomplished using the signal processing device of the invention. For example, Figure 7 illustrates an individual 61 in room 57. Also in the room is a communications device 60 such as a TV or a radio. Microphone 54 is located near the noise source 60. Microphone 56 is near the desired signal source from the individual 61. Due to the room acoustics, both signals are picked up by both microphones. The processing and reconstruction device 58 operates on the detected signals in accordance with the principles described above. Thus, the desired signal from source 61 can be derived without interfering reverberation being caused by the processing. The desired signal reconstructed from device 58 can then be processed by a speech recognition device 63.

The signal separator of the invention can also be used in a moving vehicle, such as a car. A microphone can be located near the muffler and

another microphone near the speaker. The speaker's microphone could be a car phone. The noise caused by the muffler can be eliminated by using the signal separator of the invention. Moreover, since some of the generating transfer functions are known or can be calculated from the car acoustical environment, calculations can be simplified in the reconstruction filter.

Figure 8 illustrates a signal enhancing system in accordance with the invention in an underwater acoustic environment. A sonar array 64 is used to determine the position of target 66. Noise generated by propeller 70 interferes with this determination. Accordingly, microphone 62 generates an output signal which is processed with the signals from array 64 by signal separation device 68. Thus, despite the leakage of sonar signals from array 64 to microphone 62, an accurate target signal can be calculated.

Although speech processing is the preferred application of the invention, the signal processing device can be applied in many fields. For example, noise cancellation in a telemetry system can be effected by means of the signal enhancement device. Electrical signals from the human body, such as those processed in an electrocardiogram, can be more accurately determined by means of the invention. Also, antenna signals can be more accurately determined by means of the invention.

Equivalents

Those skilled in the art will recognize, or be able to ascertain, using no more than routine experimentation, many equivalents to the specific embodiments of the invention described herein.

These and all other equivalents are intended to be encompassed by the following claims.

CLAIMS

1. A signal processing system comprising:

a plurality of detectors for detecting source signals subjected to a generating system transfer function;

a processor for receiving the detected signals and producing a reconstruction filter for filtering the detected signal to produce reconstructed source signals, the reconstruction filter being produced such that correlation between reconstructed source signals is forced to approach zero even when each detector receives sources signals from plural sources; and

said reconstruction filter for receiving the detected signals to generate a reconstruction of the various source signals without interference from any of the other source signals.

2. A signal processing system, as recited in Claim 1, wherein the reconstruction filter is inversely related to said transfer function.

3. A signal processing system, as recited in Claim 2, wherein some of the transfer function parameters are known.

4. A signal processing system, as recited in Claim 2, wherein none of the transfer function parameters are known and the transfer function has a finite impulse response.

5. A signal processing system, as recited in Claim 4, wherein the reconstruction filter is iterative.

6. A signal processing system, as recited in Claim 4, wherein the reconstruction filter is adaptive.

7. A signal processing system, as recited in Claim 1, wherein the reconstruction filter has a finite impulse response.

8. A signal processing system comprising:

a first detector for receiving at least one source signal subjected to a generating system transfer function;

a second detector for receiving at least one source signal subjected to said generating system transfer function;

a reconstruction filter including a first transfer function to which a first signal from the first detector is applied and a second transfer function to which a second input signal from the second detector is applied, the output of the first transfer function being subtracted from the second input signal and the output of the second transfer function being subtracted from the first input signal;

such that the reconstruction filter for filtering the first and second signals produce reconstructed first and second source signals with a correlation that is forced to approach zero even when each detector receives source signals from plural sources.

9. A signal processing system, as recited in Claim 8, wherein the first and second transfer functions are tapped delay lines with a plurality of equal delay segments, the output of each delay segment is multiplied by a filter coefficient, and the filter coefficient outputs for the respective delay segments are summed to serve as outputs for the first and second transfer functions.

10. A signal processing system, as recited in Claim 9, wherein the reconstruction filter is inversely related to said transfer function.

11. A signal processing system, as recited in Claim 10, wherein some filter coefficients are known.

12. A signal processing system, as recited in Claim 10, wherein none of the filter coefficients are known and the generating system transfer function has a finite impulse response.

13. A signal processing system, wherein the reconstruction filter is iterative.

14. A signal processing system, as recited in Claim 12, wherein the reconstruction filter is adaptive.

15. A signal processing system, as recited in Claim 1, wherein the reconstruction filter has a finite impulse response.

16. A method of multichannel source signal separation, wherein said source signals are subjected to a transfer function, comprising the steps of:

detecting a plurality of observed signals from at least one source signal subjected to the transfer function; and

processing the detected signals and producing a reconstruction filter for filtering the detected signal to produce reconstructed source signals such that the correlation between the various reconstructed source signals is forced to approach zero even when each observed signal receives source signals from plural sources.

17. A method, as recited in Claim 16, wherein the reconstruction filter is inversely related to said transfer function.

18. A method, as recited in Claim 17, wherein some of the transfer function parameters are known.

19. A method, as recited in Claim 17, wherein none of the transfer function parameters are known and the transfer function has a finite impulse response.

20. A method, as recited in Claim 19, wherein the reconstruction filter is iterative.

21. A method, as recited in Claim 19, wherein the reconstruction filter is adaptive.

22. A method, as recited in Claim 16, wherein the reconstruction filter has a finite impulse response.

23. A signal processing system, for reconstructing first source signal S_1 and a second source signal S_2 which are subjected to a generating transfer function, comprising

(a) a first detector which produces a first observed signal y_1 from at least one source signal subjected to said transfer function;

(b) a second detector which produces a second observed signal y_2 from at least one source signal subjected to said transfer function;

(c) a reconstruction filter including:

(i) a first filter H_{21} which processes the first observed signal y_1 and produces a first output signal U_1,

(ii) a second filter H_{12} which processes the second observed signal y_2 and produces a second output signal U_2,

(iii) a first subtractor which subtracts V_2 from Y_1 to produce V_1,

(iv) a second subtractor which subtracts U_1 from Y_2 to produce V_2,

(v) a processing transform which converts the signals V_1 and V_2, respectively, into reconstructed source signals, S_1 and S_2, respectively, wherein the correlation between the reconstructed source signals is forced to approached zero, even when each detector receives source signals from plural sources.

24. A system, as recited in Claim 23, wherein the reconstruction filter is inversely related to said transfer function.

25. A system, as recited in Claim 24, wherein some of the transfer function parameters are known.

26. A system, as recited in Claim 24, wherein none of the transfer function parameters are known and the transfer function has a finite impulse response.

27. A system, as recited in Claim 26, wherein the reconstruction filter is iterative.

28. A system, as recited in Claim 26, wherein the reconstruction filter is adaptive.

29. A system, as recited in Claim 23, wherein the reconstruction filter has a finite impulse response.

30. A system for enhancing speech in a noise environment, where both the speech and noise are subjected to a generating system transfer function, comprising:

a first microphone near a speaker which produces a first observed signal y_1, including at least one of speech and noise components,

a second microphone near a noise source which produces a second observed signal Y_2, including at least one of speech and noise components,

a processor for receiving the first and second observed signals, y_1 and y_2, and producing a reconstruction filter for filtering the detected signals to produce reconstructed source signals,

said reconstruction filter producing separate reconstructed speech and noise signals such that the correlation between the reconstructed signals is forced to approach zero even when each microphone detects both speech and noise components..

31. A system, as recited in Claim 30, further comprising a speech recognition device coupled to said reconstruction filter.

32. A system, as recited in Claim 30, wherein the reconstruction filter is inversely related to the transfer function.

33. A system, as recited in Claim 32, wherein some of the transfer function parameters are known.

34. A system, as recited in Claim 32, wherein none of the transfer function parameters are known and the transfer function has a finite impulse response.

35. A system, as recited in Claim 34, wherein the reconstruction filter is iterative.

36. A system as recited in Claim 32, wherein the reconstruction filter is adaptive.

37. A system, as recited in Claim 30, wherein the reconstruction filter has a finite impulse response.

38. A signal enhancing system in an underwater acoustic environment, where both a desired signal and noise are subjected to a generating system transfer function, comprising

a sonar array for producing and receiving ultrasonic signals, wherein the received signals represent a plurality of first observed signals Y_1-Y_p, including at least one of desired signal and noise components,

a microphone near a noise source which produces a second observed signal Y_2, including at least one of desired signal and noise components,

a processor for receiving the first and second observed signals Y_1-Y_p and Y_2, and producing a reconstruction filter for filtering the detected signals to produce reconstructed source signals,

said reconstruction filter producing separate reconstructed signals where the correlation between the reconstructed signals is forced to approach zero, even when the sonar array and the microphone both detect desired signal and noise components.

39. A system, as recited in Claim 32, wherein the reconstruction filter is inversely related to the transfer function.

40. A system, as recited in Claim 39, wherein some of the transfer function parameters are known.

41. A system, as recited in Claim 39, wherein none of the transfer function parameters are known and the transfer function has a finite impulse response.

42. A system, as recited in Claim 41, wherein the reconstruction filter is iterative.

43. A system, as recited in Claim 41, wherein the reconstruction filter is adaptive.

44. A system, as recited in Claim 38, wherein the reconstruction filter has a finite impulse response.

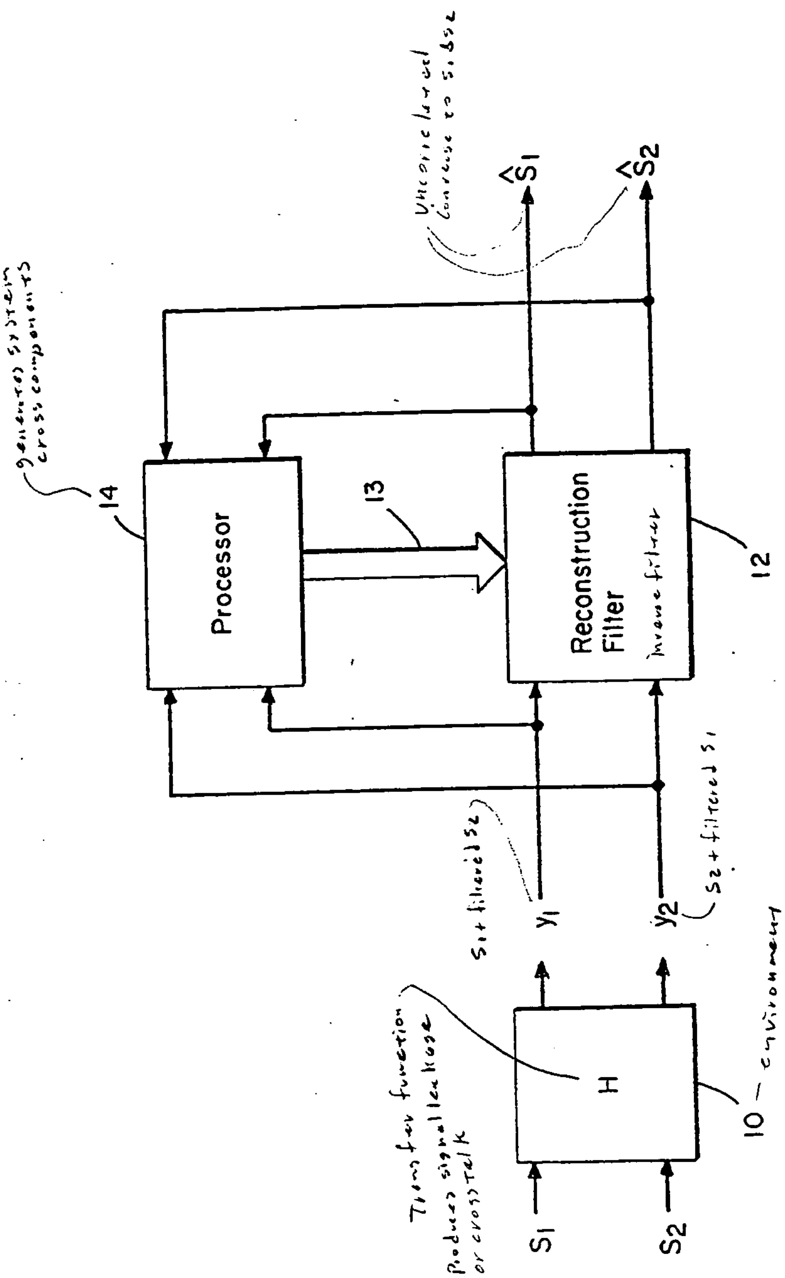

FIG. 1

Signal processing system

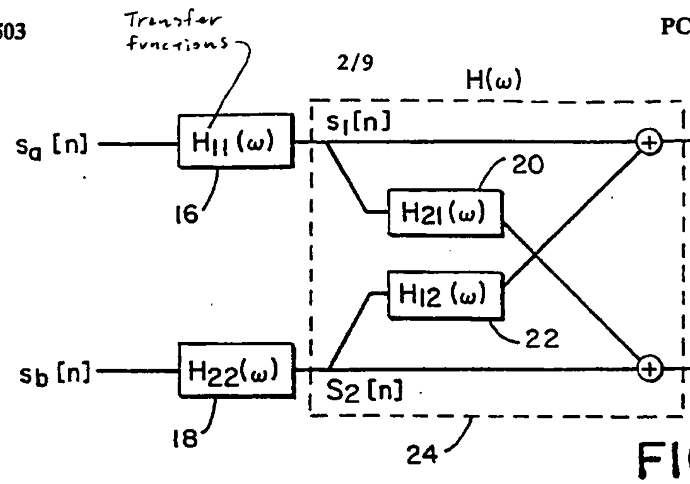

FIG. 2

linear invariant

2 channel signal processing system

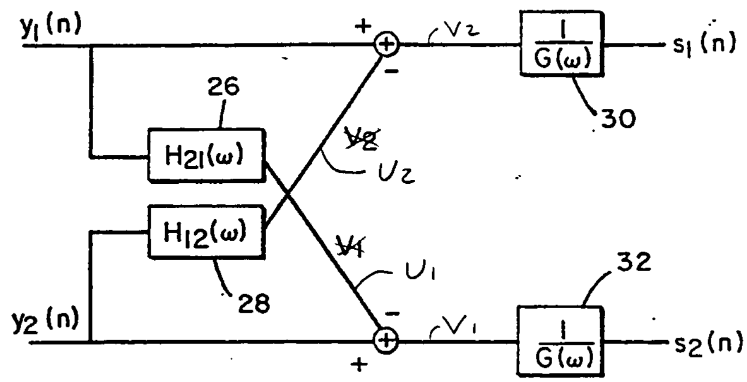

FIG. 3

2 channel signal processing sy

inverse of Fig 2

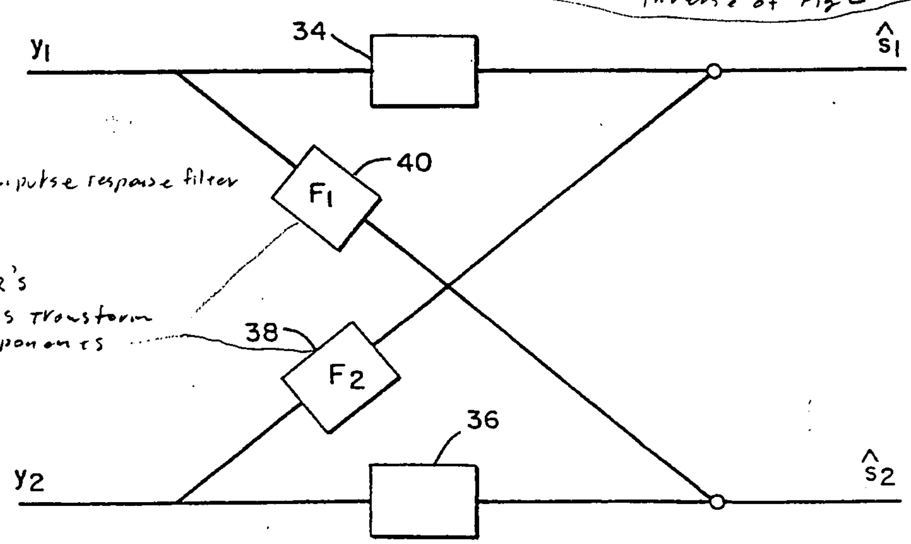

FIG. 4

Reconstruction system

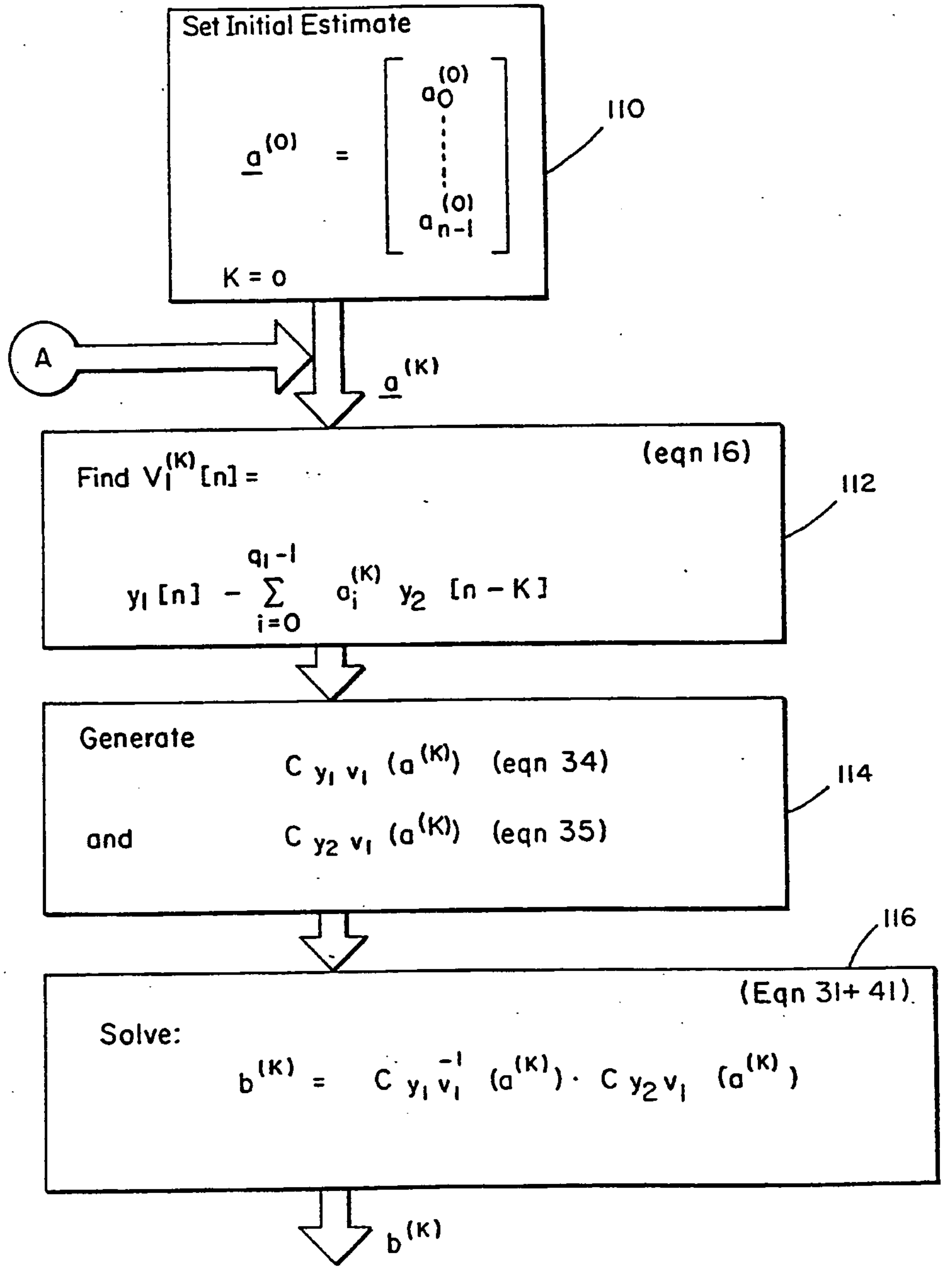

FIG. 5 SHEET I

Signal processing system
interative process

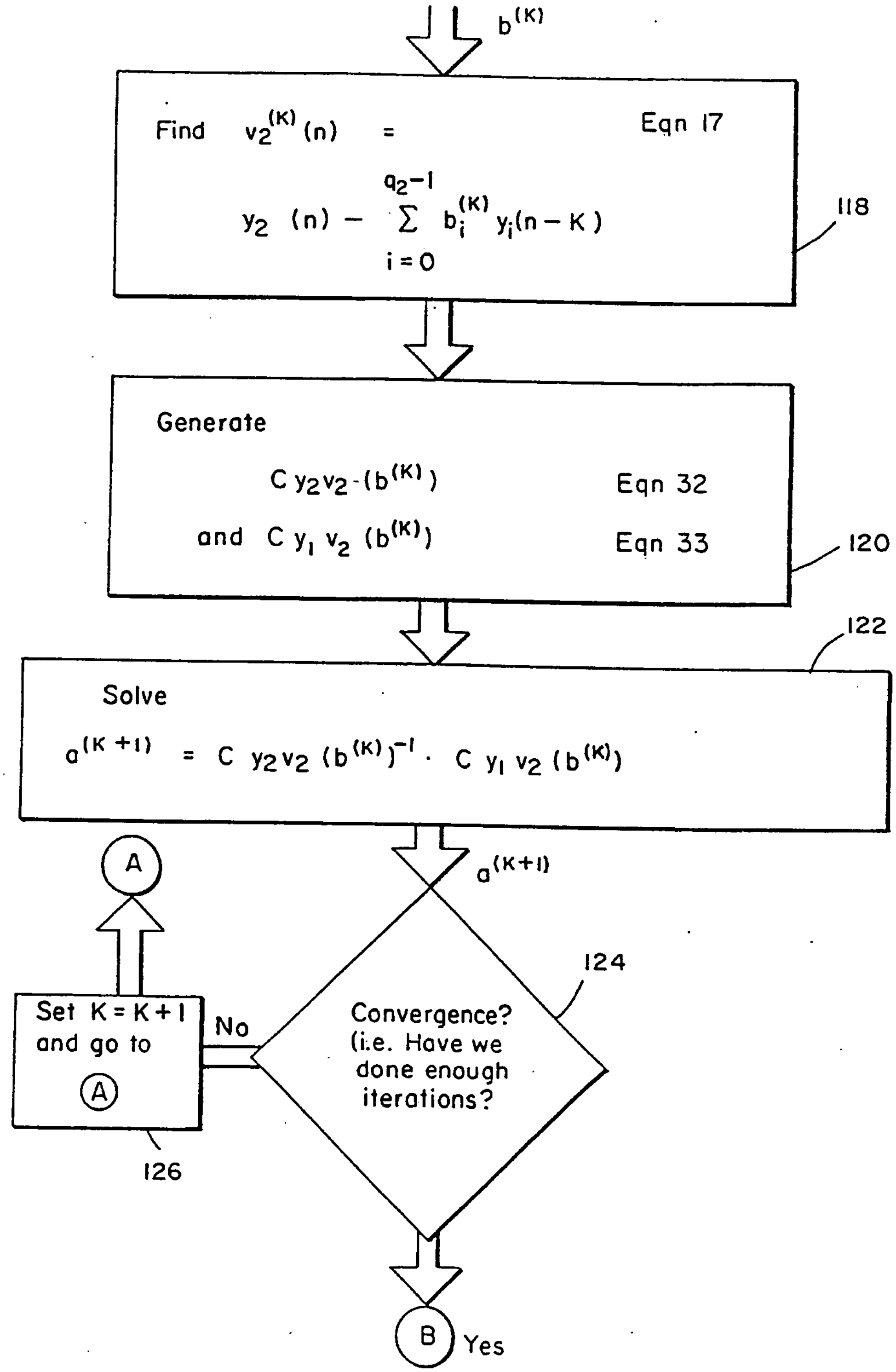
$b^{(K)}$
Find $v_2^{(K)}(n) = y_2(n) - \sum_{i=0}^{q_2-1} b_i^{(K)} y_i(n-K)$ Eqn 17
118
Generate
$C_{y_2 v_2}(b^{(K)})$ Eqn 32
and $C_{y_1 v_2}(b^{(K)})$ Eqn 33
120
122
Solve
$a^{(K+1)} = C_{y_2 v_2}(b^{(K)})^{-1} \cdot C_{y_1 v_2}(b^{(K)})$
A
$a^{(K+1)}$
124
Set K = K + 1 and go to A
No
Convergence? (i.e. Have we done enough iterations?
126
B
Yes

FIG. 5 Sheet 2

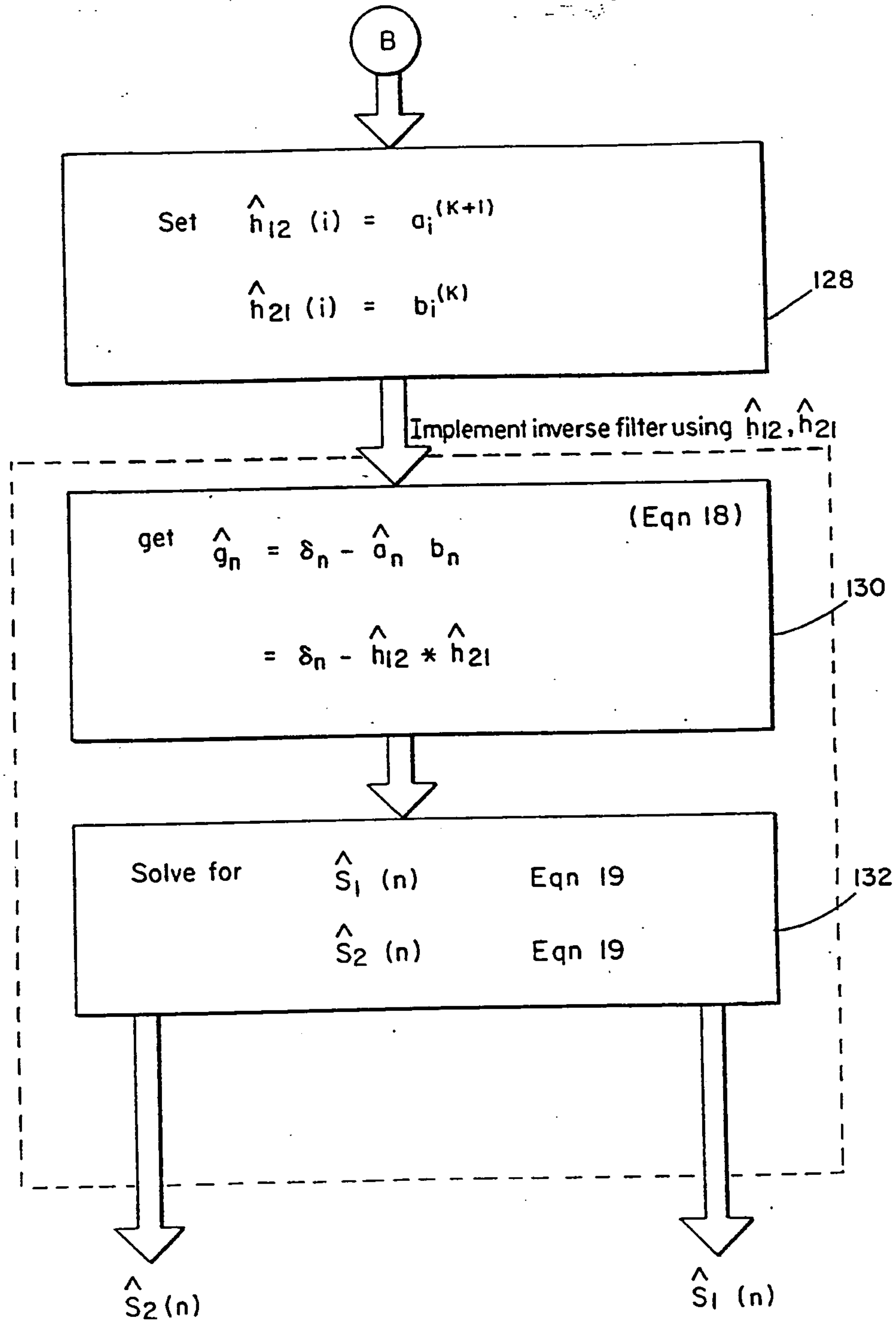
B
Set $\hat{h}_{12}(i) = a_i^{(K+1)}$
$\hat{h}_{21}(i) = b_i^{(K)}$
128
Implement inverse filter using $\hat{h}_{12}, \hat{h}_{21}$
get $\hat{g}_n = \delta_n - \hat{a}_n \; b_n$
$= \delta_n - \hat{h}_{12} * \hat{h}_{21}$
(Eqn 18)
130
Solve for $\hat{S}_1(n)$ Eqn 19
$\hat{S}_2(n)$ Eqn 19
132
$\hat{S}_2(n)$
$\hat{S}_1(n)$

FIG. 5 Sheet 3

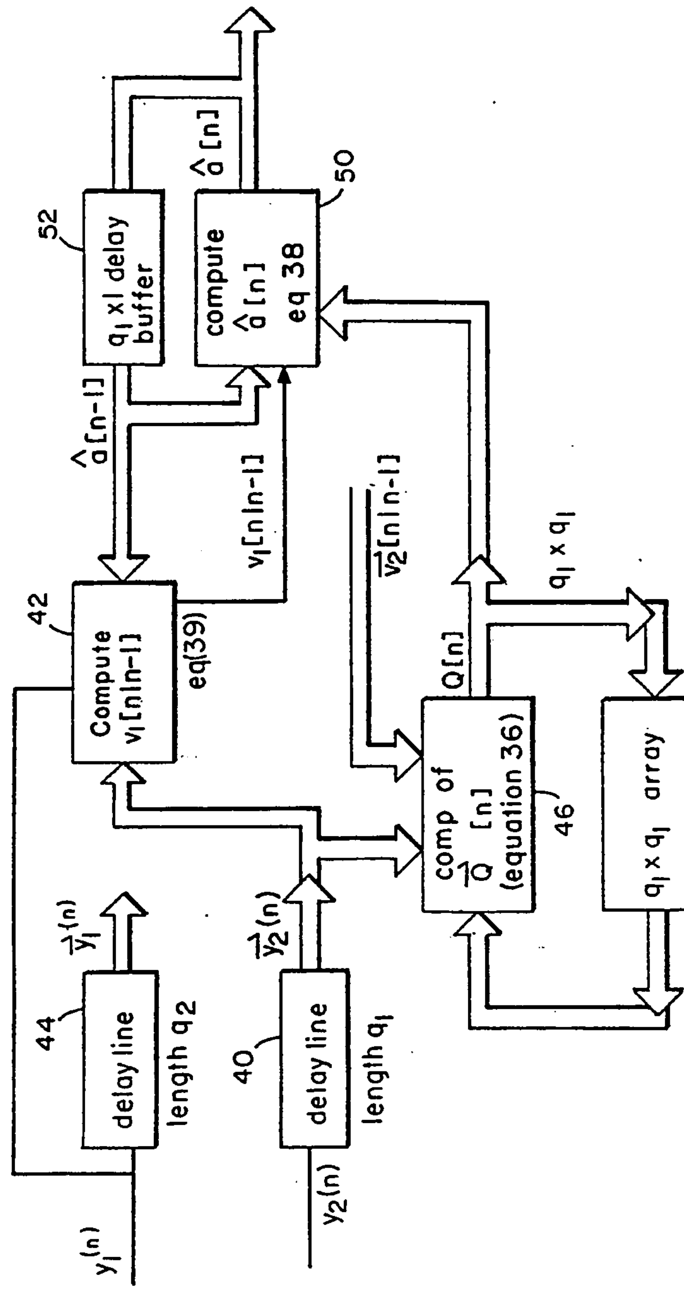

FIG. 6

Signal Processing system adaptive sequential algorithm

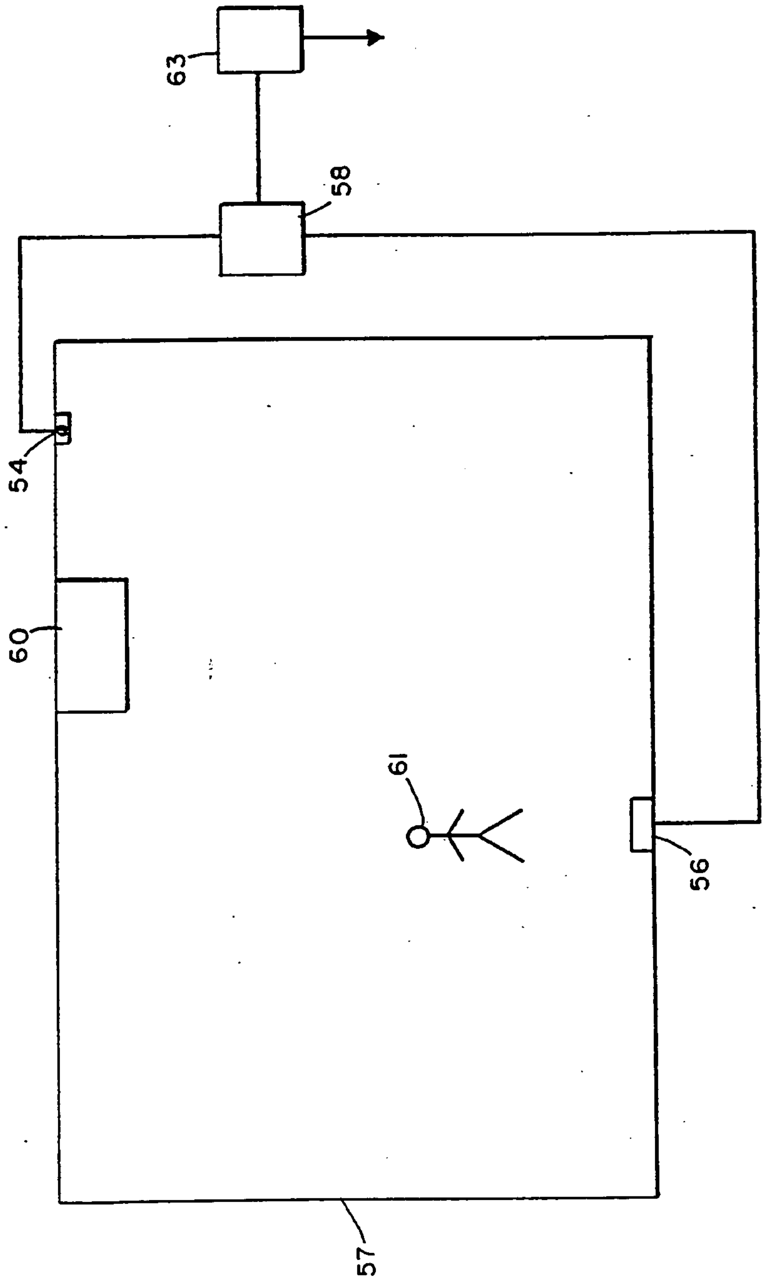

FIG. 7

speech enhancing
noise cancelling

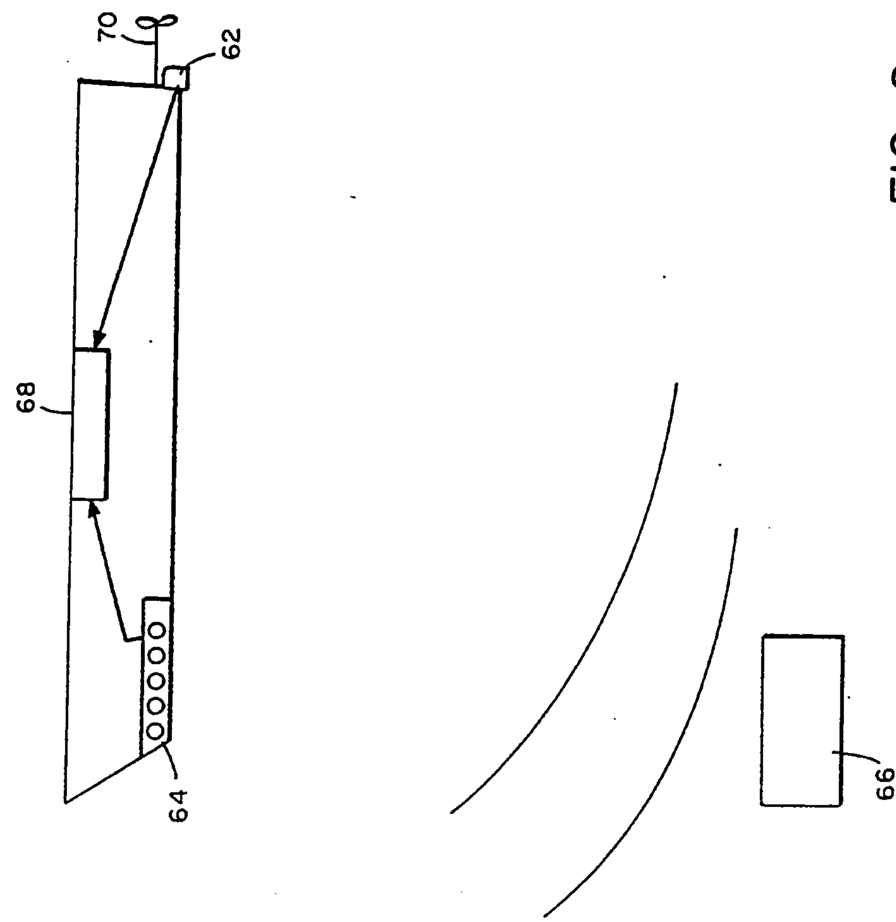

FIG. 8

Signal enhancing underwater system

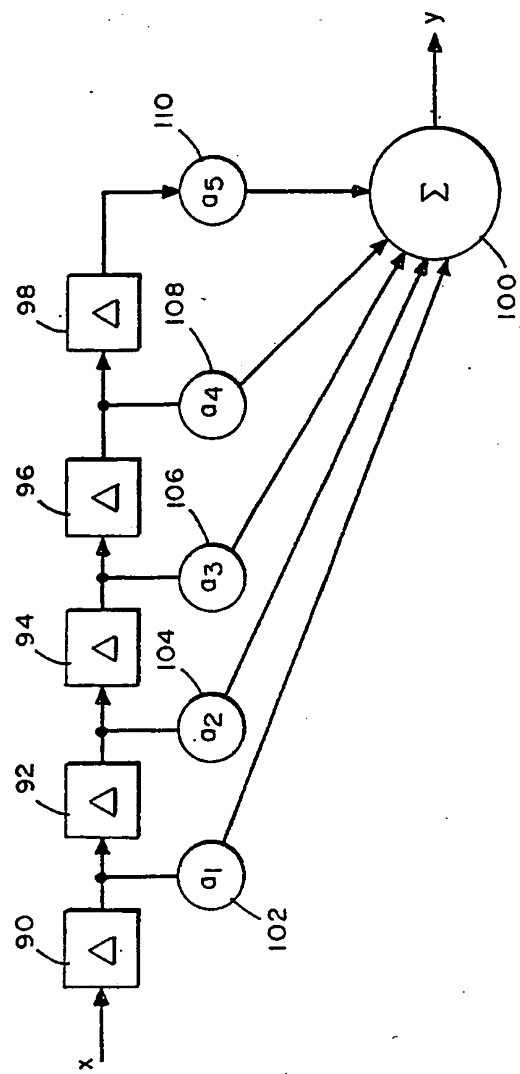

FIG. 9
Tapped delay line

ANHANG	ANNEX	ANNEXE
zum internationalen Recherchenbericht über die internationale Patentanmeldung Nr.	to the International Search Report to the International Patent Application No.	au rapport de recherche international relatif à la demande de brevet international n°

PCT/US 92/07355 SAE 64753

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Im Recherchenbericht angeführtes Patentdokument Patent document cited in search report Document de brevet cité dans le rapport de recherche	Datum der Veröffentlichung Publication date Date de publication	Mitglied(er) der Patentfamilie Patent family member(s) Membre(s) de la famille de brevets	Datum der Veröffentlichung Publication date Date de publication
WO A1 8801054	11-02-88	US A 4702110	27-10-87
		US A 4602511	29-07-86
EP A2 210609	04-02-87	BR A 8603507	04-03-87
		EP TD 210609	21-05-87
		EP A3 210609	15-03-89
		ES AF 2001168	01-05-88
		JP A2 62029232	07-02-87
		US A 4677466	30-06-87
US A 4499601		keine - none - rien	

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INTERNATIONAL SEARCH REPORT

International Application No PCT/US 92/07355

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all)[6]

According to International Patent Classification (IPC) or to both National Classification and IPC

IPC^5: G 10 L 9/08

II. FIELDS SEARCHED

Minimum Documentation Searched [7]

Classification System	Classification Symbols
IPC^5	G 10 L 9/00, G 10 L 1/00, H 04 H 9/00, H 04 N 7/00, H 04 R 3/00, H 03 J 7/00, H 04 H 1/00

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched [8]

III. DOCUMENTS CONSIDERED TO BE RELEVANT[9]

Category *	Citation of Document, [11] with indication, where appropriate, of the relevant passages [12]	Relevant to Claim No. [13]
A	WO, A1, 88/01 054 (MCS PARTNERS) 29 December 1988 (29.12.88), see fig. 1; abstract. --	1-44
A	EP, A2, 0 210 609 (A.C. NIELSEN COMP) 04 February 1987 (04.02.87), see fig. 1; abstract. --	1-44
A	US, A, 4 499 601 (MATTHEWS) 12 February 1985 (12.02.85), see fig. 1,2; abstract. ----	1-44

* Special categories of cited documents: [10]

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search	Date of Mailing of this International Search Report
21 December 1992	15 JAN 1993
International Searching Authority	**Signature of Authorized Officer**
EUROPEAN PATENT OFFICE	BERGER e.h.

Form PCT/ISA/210 (second sheet) (January 1985)

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